What is Claimed Is:

1. A method in a user interface resource configured for providing user interface services to a subscriber, the method comprising:

establishing a first Real Time Protocol (RTP) data stream for a user interface session with the subscriber according to H.323 protocol;

initiating a second RTP data stream to a destination party in response to reception of a command from the subscriber;

connecting the first and second RTP data streams in response to detecting a first prescribed condition from the destination party; and

resuming the user interface session with the user interface resource in response to detecting a second prescribed condition between the subscriber and the destination party.

2. The method of claim 1, wherein the initiating step includes:

determining a destination phone number from the command and initiating a call to the destination phone number;

storing H.245 protocol capabilities of the destination phone number as the call is initiated; and

establishing the second RTP data stream to the destination party upon termination of the call.

3. The method of claim 2, wherein the step of initiating a call to the destination phone number includes initiating the call by the user interface resource using an IP telephony gateway.

4. The method of claim 3, wherein the step of establishing the second RTP data stream includes initiating the second RTP data stream in response to detecting an off hook condition at the destination phone number.

5. The method of claim 2, wherein the determining step includes identifying the destination phone number by one of:

using speech recognition to recognize the destination phone number from the command; and

accessing a database, configured for storing telephone numbers relative to prescribed subscriber commands, for retrieval of the destination phone number based on the command.

6. The method of claim 1, wherein the step of connecting the first and second RTP data streams includes closing the first and second RTP data streams to the user interface resource by sending to an IP telephony gateway, configured for establishing the first and second RTP data streams with the subscriber and the destination party, respectively, Empty Capability Set messages across an H.245 protocol channel for the first and second RTP data streams, respectively, wherein the IP telephony gateway in response closes the first and second RTP data streams to the user interface resource.

7. The method of claim 6, wherein the step of connecting the first and second RTP data streams further includes:

resetting the IP telephony gateway to an initialized state by sending a Non-Empty Capability Set message for the first and second RTP data streams; and

connecting within the IP telephony gateway a first port servicing the first RTP data stream with a second port servicing the second RTP data stream.

8. The method of claim 7, wherein the step of connecting the first and second port includes supplying to the IP telephony gateway a first RTP port number specifying the first port for connection with the second port, and a second RTP port number specifying the second port for connection with the first port.

9. The method of claim 7, wherein the resuming step includes:

detecting a disconnect message based on the destination party disconnecting from the second RTP data stream;

sending an acknowledgment to the IP telephony gateway to clear the second RTP data stream; and

reconnecting with the first RTP data stream to resume the user interface session.

10. The method of claim 9, wherein the reconnecting step includes:

reissuing an Empty Capability Set Message to the IP telephony gateway for the first RTP data stream; and

setting up the first RTP data stream for reception by the user interface resource.

11. The method of claim 1, wherein the resuming step includes:

detecting a disconnect message based on the destination party disconnecting from the second RTP data stream;

sending an acknowledgment to an IP telephony gateway to clear the second RTP data stream; and

reconnecting with the first RTP data stream to resume the user interface session.

12. The method of claim 11, wherein the reconnecting step includes:

issuing an Empty Capability Set Message to the IP telephony gateway for the first RTP data stream; and

setting up the first RTP data stream for reception by the user interface resource.

13. A system configured for providing user interface services to a subscriber over an Internet protocol (IP) telephony link, the system comprising:

an IP telephony gateway configured for establishing Real Time Protocol (RTP) data stream connections according to H.323 protocol; and

a user interface resource configured for establishing a first RTP data stream connection with the subscriber via the IP telephony gateway for a user interface session, the user interface resource configured for initiating a second RTP data stream to a destination party for establishment of a call between the subscriber and the destination party in response to a call command from the subscriber, the user interface resource resuming the user interface session with the subscriber in response to a detected disconnect condition between the subscriber and the destination party.

14. The system of claim 13, wherein the user interface resource establishes the call by sending bridging commands to the IP telephony gateway, the IP telephony gateway in response closing the first and second RTP data streams to the user interface resource and bridging the first and second RTP data streams.

15. The system of claim 13, wherein the user interface resource outputs Empty Capability Set messages for the first and second RTP data streams to the IP telephony gateway across an H.245 channel, the IP telephony gateway in response closing the first and second RTP data streams to the user interface resource.

16. The system of claim 15, wherein the user interface resource outputs Non-Empty Capability Set messages for the first and second RTP data streams to the IP telephony gateway across the H.245 channel, the IP telephony gateway in response initiating bridging of the first and second RTP data streams.

17. The system of claim 16, wherein the IP telephony gateway initiates the bridging by sending an Open Logical Channel request to the user interface resource, the user interface resource in response sending an acknowledgment and media stream addresses for the first and second RTP data streams, the IP telephony gateway bridging the first and second RTP data streams based on the media stream addresses.

18. The system of claim 13, wherein the user interface resource, in response to detecting the disconnect condition, outputs to the IP telephony gateway an Empty Capability Set message for the first RTP data stream and an acknowledgment to clear the second RTP data stream, for reconnection of the first RTP stream with the user interface resource.

19. The system of claim 13, wherein the user interface resource determines a destination telephone number for the destination party based on recognizing speech representing the destination telephone number within the call command.

20. The system of claim 13, wherein the user interface resource determines the destination telephone number for the destination party based on retrieval from a database, configured for storing telephone numbers for prescribed destinations, using the destination party as a search key.

21. A computer readable medium having stored thereon sequences of instructions for executing user interface services by a user interface resource for a subscriber, the sequences of instructions including instructions for performing the steps of:

establishing a first Real Time Protocol (RTP) data stream for a user interface session with the subscriber according to H.323 protocol;

initiating a second RTP data stream to a destination party in response to reception of a command from the subscriber;

connecting the first and second RTP data streams in response to detecting a first prescribed condition from the destination party; and

resuming the user interface session with the user interface resource in response to detecting a second prescribed condition between the subscriber and the destination party.

22. The medium of claim 21, wherein the initiating step includes:

determining a destination phone number from the command and initiating a call to the destination phone number;

storing H.245 protocol capabilities of the destination phone number as the call is initiated; and

establishing the second RTP data stream to the destination party upon termination of the call.

23. The medium of claim 22, wherein the step of initiating a call to the destination phone number includes initiating the call by the user interface resource using an IP telephony gateway.

24. The medium of claim 23, wherein the step of establishing the second RTP data stream includes initiating the second RTP data stream in response to detecting an off hook condition at the destination phone number.

25. The medium of claim 22, wherein the determining step includes identifying the destination phone number by one of:

using speech recognition to recognize the destination phone number from the command; and

accessing a database, configured for storing telephone numbers relative to prescribed subscriber commands, for retrieval of the destination phone number based on the command.

26. The medium of claim 21, wherein the step of connecting the first and second RTP data streams includes closing the first and second RTP data streams to the user interface resource by sending to an IP telephony gateway, configured for establishing the first and second RTP data streams with the subscriber and the destination party, respectively, Empty Capability Set messages across an H.245 protocol channel for the first and second RTP data streams, respectively, wherein the IP telephony gateway in response closes the first and second RTP data streams to the user interface resource.

27. The medium of claim 26, wherein the step of connecting the first and second RTP data streams further includes:

resetting the IP telephony gateway to an initialized state by sending a Non-Empty Capability Set message for the first and second RTP data streams; and

connecting within the IP telephony gateway a first port servicing the first RTP data stream with a second port servicing the second RTP data stream.

28. The medium of claim 27, wherein the step of connecting the first and second port includes supplying to the IP telephony gateway a first RTP port number specifying the first port for connection with the second port, and a second RTP port number specifying the second port for connection with the first port.

29. The medium of claim 27, wherein the resuming step includes:

detecting a disconnect message based on the destination party disconnecting from the second RTP data stream;

sending an acknowledgment to the IP telephony gateway to clear the second RTP data stream; and

reconnecting with the first RTP data stream to resume the user interface session.

30. The medium of claim 29, wherein the reconnecting step includes:

reissuing an Empty Capability Set Message to the IP telephony gateway for the first RTP data stream; and

setting up the first RTP data stream for reception by the user interface resource.

31. The medium of claim 21, wherein the resuming step includes:

detecting a disconnect message based on the destination party disconnecting from the second RTP data stream;

sending an acknowledgment to an IP telephony gateway to clear the second RTP data stream; and

reconnecting with the first RTP data stream to resume the user interface session.

32. The medium of claim 31, wherein the reconnecting step includes:

issuing an Empty Capability Set Message to the IP telephony gateway for the first RTP data stream; and

setting up the first RTP data stream for reception by the user interface resource.

13 33. A system configured for providing user interface services to a subscriber over an Internet protocol (IP) telephony link, the system comprising:

an IP telephony gateway configured for establishing Real Time Protocol (RTP) data stream connections according to H.323 protocol; and

means for establishing a first RTP data stream connection with the subscriber via the IP telephony gateway for a user interface session, the means for establishing initiating a second RTP data stream to a destination party for establishment of a call between the subscriber and the destination party in response to a call command from the subscriber, the means for establishing resuming the user interface session with the subscriber in response to a detected disconnect condition between the subscriber and the destination party.

14 34. The system of claim 33, wherein the means for establishing establishes the call by sending bridging commands to the IP telephony gateway, the IP telephony gateway in response closing the first and second RTP data streams to the means for establishing and bridging the first and second RTP data streams.

15 35. The system of claim 33, wherein the means for establishing outputs Empty Capability Set messages for the first and second RTP data streams to the IP telephony gateway across an H.245 channel, the IP telephony gateway in response closing the first and second RTP data streams to the means for establishing.

36. The system of claim 35, wherein the means for establishing outputs Non-Empty Capability Set messages for the first and second RTP data streams to the IP telephony gateway across the H.245 channel, the IP telephony gateway in response initiating bridging of the first and second RTP data streams.

37. The system of claim 36, wherein the IP telephony gateway initiates the bridging by sending an Open Logical Channel request to the means for establishing, the means for establishing in response sending an acknowledgment and media stream addresses for the first and second RTP data streams, the IP telephony gateway bridging the first and second RTP data streams based on the media stream addresses.

38. The system of claim 33, wherein the means for establishing, in response to detecting the disconnect condition, outputs to the IP telephony gateway an Empty Capability Set message for the first RTP data stream and an acknowledgment to clear the second RTP data stream, for reconnection of the first RTP stream with the means for establishing.

39. The system of claim 33, wherein the means for establishing determines a destination telephone number for the destination party based on recognizing speech representing the destination telephone number within the call command.

40. The system of claim 33, wherein the means for establishing determines the destination telephone number for the destination party based on retrieval from a database, configured for storing telephone numbers for prescribed destinations, using the destination party as a search key.